REMARKS/ARGUMENTS

Favorable reconsideration of this application, as presently amended and in light of the following discussion, is respectfully requested.

Claims 20-24 are currently pending, Claims 20-24 having been amended. The changes and additions to the claims do not add new matter and are supported by the originally filed specification, for example, on page 7, lines 5-16; page 15, lines 8-19.

In the outstanding Office Action, Claim 24 was rejected under 35 U.S.C. §112, first paragraph, as failing to comply with the written description requirement; Claim 24 was rejected under 35 U.S.C. §101 as being directed to non-statutory subject matter; Claims 20-22 and 24 were rejected under 35 U.S.C. §103(a) as being unpatentable over Lai et al. (U.S. Pub. No. 2005/0002409, hereinafter "Lai") in view of Ofek (U.S. Patent No. 6,038,230) and Lakaniemi et al. (U.S. Pub. No. 2003/00043856, hereafter "Lakaniemi"); and Claim 23 was rejected under 35 U.S.C. §103(a) as being unpatentable over Lai in view of Ofek, Lakaniemi, and Serizawa (U.S. Pub. No. 2002/0169859).

With respect to the rejection of Claim 24 under 35 U.S.C. §112, first paragraph, the Examiner states that "Claim 24 states 'means for concatenating frame output acoustic signals outputted from the acoustic signal packet decoding means or the loss handling means and outputting the concatenated frame output acoustic signal.'" However, Claim 24 does not have this recitation at all. Applicants note that previous Claim 23 did recite the above-noted feature, therefore it is assumed that the Examiner intended to rejection Claim 23 under 35 U.S.C. §112, first paragraph instead of Claim 24. Furthermore, Applicant submits that the present amendment to Claim 23, which now recites "means for generating, as a reproduced acoustic signal, frame output acoustic signals outputted from the acoustic signal packet decoding means or the loss handling means and outputting the reproduced acoustic signal," renders this ground of rejection moot.

With respect to the rejection of Claim 24 under 35 U.S.C. §101, Applicants respectfully submit that the present amendment to Claim 24, reciting a "non-transitory computer readable recording medium," overcomes this ground of rejection. Furthermore, Applicants note that Claim 24 has been amended to comply with Director Kappos' memo of January 27, 2010, which stated that the subject matter eligibility of a computer readable medium may be secured by excluding signal based embodiments described in the specification. To this end, Applicants have adopted the language "non-transitory" as suggested in the memo to address U.S Patent and Trademark Office formalities only. More specifically, it is noted that the recitation of "non-transitory" is a limitation of the medium itself (i.e, tangible, not a signal) as opposed to a limitation on data storage persistency (e.g., RAM vs. ROM).

With respect to the rejection of Claim 20 under 35 U.S.C. §103(a), Applicants respectfully traverse this ground of rejection and submit that the present clarifying amendment overcomes this ground of rejection. Amended Claim 20 recites, *inter alia*,

> in the transmitting unit:
>
> a step of dividing an acoustic signal such as a voice or music signal into given time segments called frames to generate frame acoustic signals in association with respective frame numbers and generating, from each frame acoustic signal, acoustic signal corresponding data as data corresponding to the frame acoustic signal; and
>
> a containing step of containing, in each packet, a frame acoustic signal of a current frame, an acoustic signal corresponding data of a past frame preceding the current frame by a difference between the frame numbers of the current frame and the past frame and a delay amount control information indicating the difference and transmitting the packet; and
>
> the acoustic signal packet transmitting method further comprises:
>
> in the receiving unit:
>
> a determination step of determining at least one of a jitter state of a received packet and a loss state of a received packet; and
>
> a step of using the result of the determination made in the determination step to determine, as a targeted value

of the number of stored packets, the number of packets to be stored in the receiving buffer; and
in the transmitting unit:
a step of setting the delay amount control information to a value smaller than or equal to the targeted value of the number of stored packets which is determined at the receiving unit.

In a non-limiting example, the delay amount control information (corresponding to a difference in frame number between a current frame and a past frame containing the acoustic signal corresponding data of the current frame) is set equal to or smaller than a targeted value of the number of stored packets in the receiving buffer, and the targeted value of the number of stored packets is determined based on at least one of a jitter state and a loss state of a receiving packet. Accordingly, the delay amount control information is eventually controlled dynamically depending on the state of receiving packets. For convenience, it is assumed that a packet number corresponds to a frame number in the following explanation.

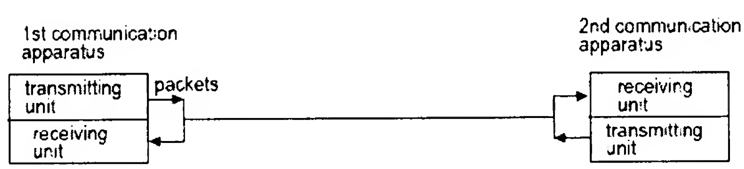
As background, Applicants submit that a scheme called "packet loss concealment" is used for transmission of an acoustic signal through a packet communication network, and there has been a known conventional scheme in which when a k-th frame failed to arrive because of a packet loss, an auxiliary information embedded in a packet after the k-th frame is used for concealment of the error.

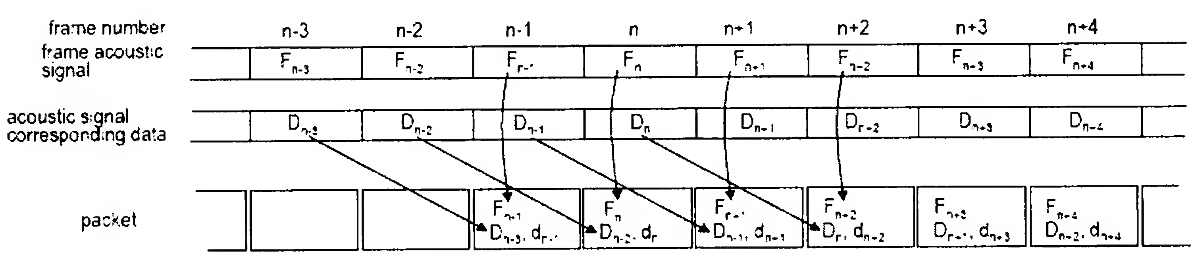
When packet loss occurs, two packets adjacent each other tend to be lost, therefore, it has been known to implement such that a frame acoustic signal and its acoustic signal corresponding data of each frame are contained respectively in two separate packets which are sufficiently apart from each other for transmission. However, as the frame number difference between a frame of a frame acoustic signal and a frame of acoustic signal corresponding data to be contained in the same packet increases, a time required for waiting an acoustic corresponding data to be used for concealment of a packet loss at the receiving side becomes longer, thus necessitating increase in delay for acoustic signal regeneration.

There have been known technologies for controlling jitter absorption buffer based on jitter of packets and/or packet loss. It has been customary to preset a fixed value for the frame number difference between the frame of "a frame acoustic signal" and a frame of "acoustic signal corresponding data" for communication between two communication apparatuses, and it has not been taken into account of the fact that an optimum frame number difference could dynamically vary depending on packet jitter and packet loss.

Amended Claim 20 clarifies that the delay amount control information corresponding to the frame number difference between the "frame acoustic signal" and the "acoustic signal corresponding data", which are to be contained in the same packet, is determined based on the receiving state of packets (i.e., at least one of a jitter state and a loss state) and, therefore, Claim 20 provides the advantage that the delay amount control information can be dynamically controlled based on the receiving state of packets.

Furthermore, for clarification purposes only and to help the Examiner understand the concept behind the claimed invention, a non-limiting example is shown below.





F_n : frame acoustic signal of n-th frame

D_n : acoustic signal corresponding data of n-th frame

d_n=n-(n-2) : delay amount control information=difference in frame number between the frame acoustic signal F_n of n-th frame and the acoustic signal corresponding data D_{n-2} of (n-2)th frame

As illustrated in the non-limiting example, a frame acoustic signal of an n-th frame, F_n, and acoustic signal corresponding data of a past frame, D_{n-dn}, which is d_n frames older than n-th frame are incorporated into the same packet together with a delay amount control information (i.e. difference value) d_n indicating a frame number difference between the frame acoustic signal of n-th frame and the acoustic signal corresponding data of d_n-older frame to be contained in the same packet. In this illustration, an example of the case where $d_n=2$ is shown. However, in the invention defined by Claim 20, the delay amount control information d_n is determined based on at least one of a jitter state and a loss state, which means the value d_n may vary depending on the jitter state or loss state.

The acoustic signal corresponding data D_n of n-th frame is produced from the frame acoustic signal F_n of the same n-th frame, and can be used at the receiving side to generate a compensating frame acoustic signal for the n-th frame when a packet containing the frame acoustic signal F_n of the n-th frame failed to arrive the receiving side. As can be seen in the above illustration, in order to obtain the acoustic signal corresponding data D_n of n-th frame, it is necessary to wait until the packet containing a frame acoustic signal F_{n+2} of the (n+2)th frame is received. Such acoustic signal corresponding data D_n may be a pitch of the frame acoustic signal of the n-th frame, or other acoustic parameters which are explained in the specification. At the receiving side, if frame acoustic signal F_n of the n-th frame is not available because of a packet loss, we can obtain the acoustic signal corresponding data D_n of the n-th frame contained in the packet which contains F_{n+2} and d_{n+2} based on the knowledge that the lost frame number $n=(n+2)-d_{n+2}$. That is, for each incoming packet, it is checked whether or not the frame number n+i (i=1, 2, ...) of the frame acoustic signal in the received packet subtracted by the difference value d_{n+i} in the same packet equals to the lost frame number n, and if yes, the acoustic signal corresponding data D_n in that packet can be used to produce a compensating frame acoustic signal for the lost acoustic signal of the n-th frame.

Turning to the applied art, Lakaniemi discloses an audio signal reproducing method in which a synchronization delay caused by a disagreement between a packet arriving rate (equal to packet transmitting rate) and a packet consumption rate required for regeneration of an acoustic signal at a receiving side is measured to determine a required amount of delay to be adjusted, and the synchronization delay is adjusted by adding/removing samples according to the delay amount. This method includes a similarity to the present invention in that a delay amount is utilized; however, the technical meaning of synchronization delay in Lakaniemi completely differs from the delay control information corresponding to the frame number difference between the frame acoustic signal and the acoustic signal corresponding data to be contained in the same packet, and the signal processing using the delay amount is also completely different between Lakaniemi and the present invention.

Lai relates to a relay system (generally called "VoIP gateway") for relaying speech data between such a synchronization network as TDM and such an asynchronization network as VoIP. Since some performances of an audio device employed in IP telephone system are similar to those of an audio device employed in the asynchronization network, jitter absorption technique employed in the speech data relay system might have been adopted from IP telephone system. However, Lai does not disclose anything about controlling a frame number difference between a frame acoustic signal and acoustic signal corresponding data to be contained in the same packet.

Ofek discloses a switching process which is capable of absorbing jitter in a switch for an asynchronization network. Such a switch is generally called "a packet shaper." Even though such information as a time stamp embedded in a header of a packet is employed in a process of absorbing jitter using a jitter memory, such a time stamp is completely different from the delay amount control information representing a frame number difference between a frame acoustic signal and acoustic signal corresponding data to be contained in the same

packet. Ofek does not disclose or suggest any idea of controlling the frame number difference in association with the control of a jitter absorption buffer.

Serizawa discloses a voice decoding method in which when some packets are received in a reverse order, a speech signal is regenerated based on packet loss concealment, while information in a memory of a decoder is replaced with the information obtainable from the late-received packet. Serizawa does not teach anything about controlling a frame number difference of a frame acoustic signal and acoustic signal corresponding data to be contained in the same packet.

Thus, Applicants submit that in Claim 20, in which a frame number difference between a frame acoustic signal and acoustic signal corresponding data to be contained in each packet is controlled based on at least one of jitter state and loss state, is not obvious based on Lai, Ofek, Lakaniemi and Serizawa, either alone or in proper combination.

Specifically, the Examiner states that paragraphs [0085] and [0086] of Lai disclose a step of dividing an acoustic signal such as a voice or music signal into given time sequence called frames to generate a frame acoustic signal, a step of generating from each acoustic signal corresponding data corresponding to the frame acoustic signal of each frame and a step of containing the frame acoustic signal and acoustic signal corresponding data in packets and transmitting the packets. The Examiner further states that paragraphs [0097], [0099], [0100], [0105] disclose a step of determining at least one of jitter state and a loss state, based on which a targeted value of the number of stored packets is determined.

However, Lai relates to a multi-channel connection device for connecting an asynchronous data network (such as the internet) with a synchronous network (such as TDM), in which time segments of voice data are received from time slots of TDM and packed into packets to be transmitted through the internet and vice versa. The time segments of voice data may be regarded as frame-divided voice signal. Therefore, the multi-channel

connection device by itself does not conduct frame-dividing of a voice signal. The cited paragraphs [0085], [0086] explain that a dual buffer 206 (Fig. 2 or 110 in Fig. 1) includes two buffers 802, 804 which are alternately read out and written in to reduce latency. Packetization of voice data is performed at VPBM (voice packet buffer memory) 204 in Fig. 2 (or Fig. 5) as suggested at paragraph [0077], lines 10-11. Thus, paragraphs [0085] and [0086] do not disclose or suggest anything about generating acoustic signal corresponding data corresponding to the frame acoustic signal.

In paragraphs [0097], [0099], [0100], [0105] of Lai, it may be disclosed that if voice data for a given channel is corrupted and/or is not received within a predetermined amount of time, a state information indicating invalid data is recorded in a JBVB memory 920, and when the JBVB memory shows a state of packet loss and/or invalid voice data, neither read address nor write address is generated for a local buffer 906. However, there is no description in Lai about determining of a targeted value of the number of stored packets based on at least one of jitter state and a loss state.

The Examiner further states that the containing step of previous Claim 20 is disclosed in Ofek at col. 4, lines 53-61; col. 9, lines 6-14; and col. 11, lines 43-56. However, Ofek relates to packet switching, wherein a virtual pipe is set along a selected route in the IP network, and at each of switches along the route a forwarding time interval is determined with reference to a common time reference signal (GPS) to reduce jitter and packet losses at the output end of the virtual pipe. In other words, in Ofek, each switch is nothing more than a repeater and does not perform frame-dividing of an acoustic signal.

In Ofek, a jitter memory (i.e., receiving buffer) is used for storing received packets whose arrival delays vary and reading out packets therefrom at equal time intervals to obtain a received signal having no variation in delay. Applicants submit that this technology has been well-known and the use of a jitter memory merely suggests the use of such technology.

That is, the jitter memory (buffer memory) is used to absorb arrival delay variation of input packets (col. 1, line 29).

Ofek discloses a technology of controlling a send-out timing of each packet using GPS as a reference at each switch along the transmission route. It is described at col. 8, lines 25-28 that each packet is given a pipe-ID and at col. 10, lines 35-36 that a header of each packet includes a time stamp. However, there is no disclosure or suggestion of containing delay amount control information in each packet. The time frame delimitter (TFD) used in Ofek is sent separately of packets each time a CTR (common time reference, e.g. GPS) is generated, as explained at col. 14, lines 45-53. TFD is not supposed to be contained in a packet. In cols. 4, 9 and 11 cited by the Examiner, there is disclosed nothing similar to a delay amount control information to be contained in each packet.

As described at col. 8, lines 50-60 of Ofek "[a] common time reference signal (such as GPS) is coupled to each of the switches, and a time assignment controller assigns selected predefined frames for transfer into and out from each of the respective switches," a predetermined time frame is used for input and output at each switch.

The Examiner acknowledges that neither Lai nor Ofek discloses setting of delay amount control information to a targeted number of stored packets or smaller, and the Examiner relies on paragraphs [0012] and [0025] of Lakaniemi to remedy this deficiency.

Lakaniemi relates to a device for regenerating an audio signal from packets received from a network, wherein synchronization delay is adjusted by insertion or removal of samples. This regeneration device does not possess either functions or constructions for transmitting an acoustic signal in packets. Therefore, Lakaniemi does not disclose or suggest anything about setting delay amount control information to a targeted value.

Therefore, Applicants respectfully submit that amended Claim 20 (and all associated dependent claims) patentably distinguishes over Lai, Ofek, Lakaniemi and Serizawa, either alone or in proper combination.

Furthermore, the Examiner had rejected Claim 21 in a similar manner as Claim 20. Additionally, Claim 21 recites "a step of sending the targeted value of the number of stored packets to the transmitting unit in the second communication apparatus," which the Examiner believes is disclosed in paragraphs [0092] and [0095] of Lai. Claim 21 also recites "a step of containing the targeted value of the number of stored packets .... in a packet as information for specifying delay amount control information to be set in the transmitting unit in the first communication apparatus," which the Examiner believes is disclosed in paragraphs [0040] and [0097] of Lai.

Applicants respectfully disagree with this assertion. Lai's paragraphs [0092] and [0095] describe of an address generator for generating write addresses for writing received packets in a jitter buffer 918 and read addresses for reading out data from the jitter buffer. Paragraph [0093] of Lai explains that the data read out of the jitter buffer 918 is provided to a local buffer 906. This is the same as in Fig. 1, where data read out of the jitter buffer 108 is written in a local buffer 110, but there is no suggestion of the step of sending the targeted value recited in claim 21.

Lai's paragraph [0040] describes that egress 112 transfers voice data from a local buffer 110 to VPBM (voice packet buffer memory) 106, and ingress 114 transfers voice data from the jitter buffer 108 to the local buffer 110. Lai's paragraph [0097] explains with reference to Fig. 9 that data of a certain voice channel is stored in the jitter buffer 918 at a location determined by a time stamp.

As already discussed above, the step of setting delay amount control information recited in Claim 21 is not disclosed in Lakaniemi for similar reasons as discussed above for

Claim 20. It is noted that in Claim 20, the receiving unit in a communication apparatus determines a targeted value, and the transmitting unit in the same communication apparatus sets the delay amount control information to the targeted value or smaller, while in Claim 21, the step of sending the targeted value and the step of containing the targeted value both define processes performed in the second communication apparatus and the step of setting delay amount control information to a value smaller than or equal to the targeted value is performed in the first communication apparatus. Those features of Claim 21 are not disclosed in Lakaniemi.

Therefore, Applicants submit that the Office Action fails to show how the applied art discloses or suggests all of the features of Claim 21 for the additional reasons set forth above.

Regarding Claim 22, the Examiner had rejected Claim 22 in a similar manner as Claim 21. Additionally, the Examiner states that Lai's paragraphs [0072] and [0076] disclose "measuring, as a remaining buffer amount, the number of packets stored in the receiving buffer." The Examiner also believes that Lai's paragraphs [0077], [0088] and [0095] disclose "containing the remaining buffer amount sent from the receiving unit in the second communication apparatus in a packet as information for specifying delay amount control information to be set in the transmitting unit in the first communication apparatus and transmitting the packet...setting delay amount control information to the remaining buffer amount contained in a packet sent from the transmitting unit in the second communication apparatus."

Applicants respectfully disagree with this assertion. Lai's paragraph [0072] discloses that VPBM 500 possesses entry groups of 502-0 to 502-z, each entry group stores a data block of one voice channel, and each entry group possesses entry 0 to entry x. Entry number "x" represents an available buffer size (memory size) for each channel, but does not mean a current number of packets remaining in the entry group. Paragraph [0076] of Lai describes

selecting a buffer size based on the number of bits of time stamp count. For example, for a time stamp of at least 8 bits, it is possible to designate an entry using 5 bits for a buffer size of 16ms. However, there is no description of determining a current number of stored packets.

Paragraph [0077] of Lai explains that upon reception of local buffer addresses and corresponding VPBM addresses, the processing engine (214 in Fig. 2 or 914 in Fig. 9) may access VPBM to obtain data from a local buffer. There is no description in Lai about a remaining buffer amount (number of packets stored in the receiving buffer), nor is there any suggestion of containing, in a packet, the remaining buffer amount as the delay amount control information. Moreover, the remaining buffer amount which the second communication apparatus contains in a packet is such one that is to be set in the transmitting unit of the first communication apparatus. However, paragraph [0077] of Lai does not teach such idea of sending the delay amount control information determined by one communication apparatus to a counter part communication apparatus to be set therein.

Moreover, Lai's paragraph [0088] describes composing a local buffer (206 in Fig. 2)of two dual port memories 802 and 804 to achieve rapid write-in. Lai's paragraph [0095] describes the address generator which receives address data from registers 910-0, 910-2 in Fig. 9. However, nether paragraph discloses anything about containing the number of packets stored in the jitter buffer 918 into a packet to be transferred from the local buffer 906 to another communication apparatus. The processing system 901 in Fig. 9 of Lai is supposed to be included in ingress 114 in Fig. 1 and generates data to be transmitted through time slots in the synchronous network (TDM). As mentioned previously, the data to be sent out through the asynchronous network (the internet) is packetized in VPBM 204 in Fig. 2 as explained in paragraph [0077].

Therefore, Applicants submit that the Office Action fails to show how the applied art discloses or suggests all of the features of Claim 22 for the additional reasons set forth above.

Regarding Claim 23, the Examiner had rejected Claim 23 in a similar manner as Claim 20. Additionally, the Examiner cites to Serizawa and states that the claimed "loss detecting means" is disclosed in paragraph [0013] of Serizawa, the claimed "acoustic signal packet decoding means" is disclosed in paragraph [0048] of Serizawa, and the "loss handling means" and :means for concatenating" is disclosed in paragraphs [0093], [0094], [0097] and the Abstract of Serizawa.

Applicants respectfully disagree with this assertion. Serizawa's paragraph [0013] explains a loss detection circuit 25 in Fig. 7, and describes that when a packet loss is detected, decoding is performed using information obtained from a packet which has already been received. Paragraph [0048] of Serizawa discloses determining whether or not a packet has been lost and conducting first filtering processing using a pitch period decoded from the received packet. However, Serizawa does not suggest anything about containing delay amount control information in a packet.

Paragraph [0094] of Serizawa describes detecting a packet loss and providing the detection result to each part, and paragraph [0097] discloses obtaining, in response to detection of a packet loss, a generation time of the packet loss from a packet in the reception buffer 10 and recording the generation time, where in case of agreement between the generation time and a generation time of a received packet, the excitation code buffer circuit 40 accumulates codes of a voice signal and a pitch filter until a time corresponding to a fixed number of packets has elapsed. Upon reception of a recalculation command from the reuse packet detection circuit 30, the excitation code buffer 40 transfers those codes accumulated after the accumulated loss packet generation time to the past excitation signal generation circuit 60. However, Serizawa does not disclose anything about extracting acoustic signal corresponding data corresponding to a frame acoustic signal from a different packet (i.e., a packet different in number by the number indicated by the delay amount control information).

Therefore, Applicants submit that the Office Action fails to show how the applied art discloses or suggests all of the features of Claim 23 for the additional reasons set forth above.

Therefore, for all of the above reasons, Applicants respectfully submit that amended Claims 20-23 (and all associated dependent claims) patentably distinguish over Lai, Ofek, Lakaniemi and Serizawa, either alone or in proper combination.

Consequently, in light of the above discussion and in view of the present amendment, the outstanding grounds for rejection are believed to have been overcome. The present application is believed to be in condition for formal allowance. An early and favorable action to that effect is respectfully requested. Furthermore, the examiner is kindly invited to contact the Applicants' undersigned representative at the phone number below to resolve any outstanding issues.

Respectfully submitted,

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